DETAILED ACTION

Continued Examination Under 37 CFR 1.114

1.   A request for continued examination under 37 CFR 1.114, including the fee set

forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this

application is eligible for continued examination under 37 CFR 1.114, and the fee set

forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action

has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on

11/22/10 has been entered.

Claim Rejections - 35 USC § 112

2.   The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly
> claiming the subject matter which the applicant regards as his invention.

3.   **Claim 1** is rejected under 35 U.S.C. 112, second paragraph, as being indefinite

for failing to particularly point out and distinctly claim the subject matter which applicant

regards as the invention.

        Claim 1 recites "A method for determining whether to accept a new call to be

routed from **the first gateway** to a second gateway in an IP network". There is

insufficient antecedent basis for "the first gateway" in the claim.

        For purpose of continuation of the prosecution, the claims will be interpreted as

best understood.

4.   The following is a quotation of the first paragraph of 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of
> making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the
> art to which it pertains, or with which it is most nearly connected, to make and use the same and shall
> set forth the best mode contemplated by the inventor of carrying out his invention.

5.   **Claims 1-33** are rejected under 35 U.S.C. 112, first paragraph, as failing to

comply with the written description requirement. The claim(s) contains subject matter

which was not described in the specification in such a way as to enable one skilled in

the art to which it pertains, or with which it is most nearly connected, to make and/or use

the invention.

Claim 1 recites the limitation "each of said network paths being associated with

respective first gateway egress interfaces and a second gateway system IP address".

There is insufficient support in the specification for this limitation in the claim.

Note that Applicant stated during telephone interview that the support of this

limitation is in page 6, line 18-31, which reads as follows:

> Packets traveling to a destination gateway can follow different paths based on the port 206x
> chosen for the specific RTP flow. The MBCAC algorithm assumes that the selection of a port 206x for an
> incoming call request is under the control of a call controller in the gateway. Hence, the MBCAC algorithm
> keeps separate admission policies for the paths from different ports to the same destination gateway. It is
> also assumed that multiple calls going from a particular port to the same destination gateway follows the
> same path, i.e., there is no load balancing within the network other than 25 provided by the gateways
> through the selection of *an egress port*. This assumption can be satisfied if the gateways use the system
> *IP address of the destination gateway* as opposed to the IP addresses of its ports. In this framework, load
> balancing is supported by controlling *the egress port at the source gateway* (i.e., first gateway 114). Since
> each egress port would map into a unique path in the IP network 118, the load from source gateway 114
> to a destination gateway (i.e., second gateway 116) can be partitioned into different paths, resulting in
> load sharing in the network

However, the above cited text does not disclose "first gateway egress interfaces"

in the claim language.

Claims 2-33 are rejected because they depend from claim 1.

Claim 31 recites the limitation "using all of the network congestion parameters".

There is insufficient support in the specification for this limitation in the claim.

Note that Applicant argues the support is in "on Pg. 9, Lines 13 - 16 of Applicants'

application, which discusses an admission control algorithm that is used to determine if there is an

uncongested path from the first gateway to the second gateway (i.e., looking at more than one, and

possibly all, of the paths from the first gateway to the second gateway)."

Page line 13-16 reads "Next, first gateway 114 consults with the admission control algorithm

to see if there is a path to the second gateway 116 that is not congested. If first gateway 114 is

15 unable to find an uncongested path to second gateway 116 it sends an error message at step 414 to

the soft switch 112..."

The cited text does not disclose "all of the network congestion parameters".

Claim 32 has the same problem because it depends from claim 31.

For purpose of continuation of the prosecution, the claims will be interpreted as

best understood.

Claim Rejections - 35 USC § 103

6.   The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all

obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set
> forth in section 102 of this title, if the differences between the subject matter sought to be patented and
> the prior art are such that the subject matter as a whole would have been obvious at the time the
> invention was made to a person having ordinary skill in the art to which said subject matter pertains.
> Patentability shall not be negatived by the manner in which the invention was made.

7.   **Claims 1, 3, 5, 14-15, 18, 23, 26** and **33** are rejected under 35 U.S.C. 103(a) as

being unpatentable over Elliott et al (US 20040022237, hereinafter Elliott) in view of

Szabo (US 20020003779 A1).

For **claim 1**, Elliott discloses a method for determining whether to accept a new

call (a call between phone 102 and phone 120, FIG. 1) to be routed from a first gateway

(gateway 108 of FIG. 1 or 21B; or SS7 Gateway 208 of FIG. 2A) to a second gateway

(gateway 110 of FIG. 1 or 21B; or SS7 Gateway 208 of FIG. 2A) in an IP network (VoIP,

[0453] and FIG. 1), comprising the steps of:

obtaining, at the first gateway, information indicative of to the quality of service

("latency or packet loss", [0099] in view of Fig. 21B) of voice calls being transmitted from

the first gateway to a second gateway via a plurality of network paths between the first

gateway and the second gateway (data network 112 of FIG. 1 or 21B that has a plurality

of network paths between the two gateways);

determining, using at least a portion of said information, a plurality of congestion

status parameters indicative of respective congestion statuses of the network paths

("latency or packet loss", [0099] in view of Fig. 21B); and

Elliott does not specifically disclose determining, using at least one of the

congestion status parameters, whether to accept the new call into the network at the

first gateway for transmission toward the second gateway via one of the network paths,

each of said network paths being associated with respective first gateway egress

interfaces and a second gateway system IP address.

In the same field of endeavor, Szabo discloses determining, using at least one of

the congestion status parameters, whether to accept the new call into the network at the

first gateway for transmission toward the second gateway via one of the network paths

("at least one *performance indicator* value read from the RTCP mechanism does not

exceed a pre-set threshold value, the call is accepted", [0028]), each of said network

paths being associated with respective first gateway egress interfaces and a second

gateway system IP address (suggested by "a method is described in B. Thompson et al.

"Tunnelling Multiplexed Compressed RTP", Internet Draft, March 2000, Work in

Progress, wherein a multitude of RTP/UDP/IP packets are compressed and multiplexed

into a so-called PPP packet", [0005]; note that a UDP/IP tunnel is a path be associated

with the IP address and egress port of each of two end points of the tunnel).

Therefore, it would have been obvious to a person of ordinary skill in the art at

the time of the invention to apply Szabo's teaching above to the gateways disclosed by

Elliott for the benefit of ensuring "a transmission path with acceptable transmission

quality" ([0008] of Szabo).

For **claim 14**, Elliott discloses an apparatus (soft switch 204 in FIG. 2 and 3, with

circuit being the means for implementing logic in the soft switch) comprising a first

gateway for interfacing voice call data from a public switch telephone network to an

Internet Protocol network; said first gateway further comprising:

for determining whether to accept a new call to be routed from a first <u>gateway</u>

(gateway 108 of FIG. 1 or 21B; or SS7 Gateway 208 of FIG. 2A) to a second <u>gateway</u>

(gateway 110 of FIG. 1 or 21B; or SS7 Gateway 208 of FIG. 2A) in an IP network (VoIP,

[0453] and FIG. 1), comprising the steps of:

a first circuit for passing the voice call data to the internet protocol network (the

Soft Switch 204 in FIG. 2B);

a second circuit for polling the internet protocol network about traffic information

transmitted therein (the Soft Switch 304 in FIG. 2B); and

a third circuit for: calculating, based on the received quality-of-service

information, a plurality of congestion status parameters associated with the respective

network paths between the first gateway and the second gateway, wherein the

congestions status parameters are indicative of respective congestion statuses of the

network paths (the circuit in the first gateway responsible for calculating quality-of-

service information such as ("latency or packet loss", [0099]); and

obtaining, at the first gateway, information indicative of to the quality of service

("latency or packet loss", [0099] in view of Fig. 21B) of voice calls being transmitted from

the first gateway to a second gateway via a plurality of network paths between the first

gateway and the second gateway (data network 112 of FIG. 1 or 21B that has a plurality

of network paths between the two gateways); and

determining, using at least a one of the congestion status parameters ("latency or

packet loss", [0099] in view of Fig. 21B).

Elliott does not specifically disclose determining, using at least one of the

congestion status parameters, whether to accept the new voice call into the network at

the first circuit for transmission toward the second gateway via one of the network paths

In the same field of endeavor, Szabo discloses determining, using at least one of

the congestion status parameters, whether to accept the new call into the network at the

first gateway for transmission toward the second gateway via one of the network paths

("at least one *performance indicator* value read from the RTCP mechanism does **not**

exceed a pre-set **threshold** value, the **call is accepted**", [0028]).

Therefore, it would have been obvious to a person of ordinary skill in the art at

the time of the invention to apply Szabo's teaching above to the gateways disclosed by

Elliott when determining whether to accept a new call or not to ensure the new call will function properly and QoS condition of the network is satisfied.

As to **claim 23**, Elliott in view of Szabo claim 1, Elliott is silent on accepting the new call into the IP network at the first gateway for transmission toward the second gateway via one of the network paths, wherein the new call is accepted into the IP network in the case of the congestion status parameter associated with the one of network paths not exceeding an upper threshold.

In the same field of endeavor, Szabo discloses accepting the new call into the IP network in the case of said parameter not exceeding an upper threshold ("at least one *performance indicator* value read from the RTCP mechanism does **not exceed** a preset **threshold** value, the **call is accepted**", [0028]; note that RTCP is a protocol for the IP network).

Therefore, it would have been obvious to a person of ordinary skill in the art at the time of the invention to apply Szabo's teaching above to the gateways disclosed by Elliott when determining whether to accept a new call or not according to KSR to ensure the new call will function properly and QoS condition of the network is satisfied.

As to **claim 3**, Elliott in view of Szabo discloses the method of claim 23, Elliott is silent on said new call is not accepted into the IP network in the case of <u>the congestion status</u> parameter <u>associated with the one of the network paths</u> exceeding the upper threshold.

Szabo further discloses said new call is not accepted into the IP network in the case of the congestion status parameter associated with the one of the network paths

exceeding the upper threshold ("one performance indicator value **exceeds** a pre-set

threshold value, the IP telephony gateway **rejects** the **call** in a step 209", [0028]; note

that the performance indicator is a congestion status parameter). The motivation of

combination is the same.

Therefore, it would have been obvious to a person of ordinary skill in the art at

the time of the invention to reject a new call in order to reduce the network congestion.

As to **claim 5**, Elliott in view of Szabo discloses the method of claim 1, Elliott

further discloses wherein the obtained information comprises, for each of at least one of

the network paths, a delay of received packets transmitted from the first gateway to the

second gateway via the network path (unacceptable latency, [1493], line 4).

As to **claim 15**, Elliott in view of Szabo discloses claim 14, Elliott further

discloses said first circuit further comprises one or more Ethernet cards ("Soft switches

204a, 204b and 204c are connected … via redundant Ethernet switches", [0568], which

suggest Ethernet interfaces with software switches as the first circuit) that are

connected to the Internet protocol network.

As to **claim 18**, Elliott in view of Szabo discloses claim 14, Elliott further

discloses the third circuit determining whether the new voice call is to be accepted into

the internet protocol network via the first circuit, by comparing each of at least one of the

congestion status parameter to at least one thresholds (Packet Loss Threshold, Table

147 – continued, page 85, where packet loss values can be used as the thresholds for

determining if the new voice call is to be accepted or not).

As to **claim 26**, Elliott in view of Szabo discloses claim 1, wherein determining whether to accept the new call into the network at the first gateway is made using call control logic (as discloses in the parent claim 1), wherein the call control logic is updated using the congestion status parameters (this limitation is the same as the last paragraph of claim 1 in that the call control logic is the one that the determining is based on).

As to **claim 33**, Elliott in view of Szabo claim 1, wherein determining whether to accept the new call into the network at the first gateway comprises:

for each of at least one of the network paths, updating a call admission control policy for the network path based on the congestion status parameter determined for the network path; and determining whether to accept the new call into the network at the first gateway based on the updated call admission control policy (as disclosed by the parent claims 1; note accepting the new call "using at least one of congestion status parameter" is updating a call admission control policy for the network path based on the congestion status parameter).

8.   **Claims 4, 6-10, 19-22, 24-25** and **27-32** are rejected under 35 U.S.C. 103(a) as being unpatentable over Elliott in view of Szabo, further in view of Schulzrinne et al., ITEF RFC 3550, July 2003 (hereinafter RFC 3550).

As to **claim 4**, Elliott in view of Szabo discloses the method of claim 1, but is silent on the information obtained is a number of send packets to said second location via the network path, wherein the number of sent packets comprises a number of lost packets, a number of late packets and a number of received packets.

RFC 3550 discloses the information obtained is a number of packets sent to said second location via the network path (Line 1 of the paragraph for "Sequence number: 16 bits", Page 14, where lost packets are those with missing sequence number), wherein the number of sent packets comprises a number of lost packets, a number of late packets (Line 1 of the paragraph for "Sequence number: 16 bits", Page 14, where late packets inherently are packets that have been sent, but have not been received according to their sequence numbers) and a number of received packets (sequence number provides information on a number of received packets).

Therefore, it would have been obvious to a person of ordinary skill in the art at the time of the invention to apply the teaching of RFC 3550 above to the system disclosed by Elliott in view of Szabo for the benefit of getting detailed information regarding network operation.

As to **claim 6**, Elliott in view of Szabo discloses the method of claim 1, but is silent on the information obtained is a delay variation (variation in the delay, Line 5 of last paragraph in Page 44) of received packets transmitted from said first location to said second location via the network path.

RFC 3550 further discloses wherein the information obtained is a delay variation (variation in the delay in RTCP packet, last paragraph in Page 44) of received packets transmitted from said first location to said second location via the network path.

Therefore, it would have been obvious to a person of ordinary skill in the art at the time of the invention to apply the teaching of RFC 3550 above to the system

disclosed by Elliott in view of Szabo for the benefit of getting detailed information regarding network operation.

As to **claim 7**, Elliott in view of Szabo discloses the method of claim 1, but is silent on the information is obtained on a periodic basis

RFC 3550 further discloses the information is obtained on a periodic basis ("based on periodic transmission", first paragraph in Page 19).

Therefore, it would have been obvious to a person of ordinary skill in the art at the time of the invention to apply the teaching of RFC 3550 above to the system disclosed by Elliott in view of Szabo for the benefit of getting detailed information regarding network operation.

As to **claim 8**, Elliott in view of Szabo discloses the method of claim 1, but is silent on the information is obtained on an exception basis using an immediate report.

RFC 3550 further discloses the information is obtained on an exception basis using an immediate report (Receiver report, first line of Section 6.4 in Page 35). The motivation of combination is the same as described in the parent claim above.

Therefore, it would have been obvious to a person of ordinary skill in the art at the time of the invention to apply the teaching of RFC 3550 above to the system disclosed by Elliott in view of Szabo for the benefit of getting detailed information regarding network operation.

As to **claim 9**, Elliott in view of RFC 3550 and Szabo discloses the method of claim 1, but is silent on the parameter including packet lost ratio.

RFC 3550 further discloses wherein the parameter include packet lost ratio (packet lost ratio, Line 1 of 3rd paragraph of Section 6.4.4, Page 43).

Therefore, it would have been obvious to a person of ordinary skill in the art at the time of the invention to apply the teaching of RFC 3550 above to the system disclosed by Elliott in view of Szabo for the benefit of getting detailed information regarding network operation.

As to **claim 10** and **19**, Elliott in view of Szabo discloses 1 and 14, but is silent on wherein PLR is defined as

$$PLR = \frac{(\text{lost packets} + \text{late packets})}{(\text{received packets} + \text{lost packets} + \text{late packets})}.$$

RFC 3550 discloses, by definition, PLR is the *ratio* of the number of *packets NOT received* to number of the *total packets sent* for a given period of time (such as disclosed by "The *ratio of these two* is the packet loss fraction over the interval" in last paragraph of page 43 in RFC 3550); The PLR formula above is simply a math expression of the definition; note that the number of packets that are NOT received equal to the *sum* of the number of lost packets and the number of last packets (*last packets* are the packets that are not received during the time interval but may be received at a later time).

Therefore, it would have been obvious to a person of ordinary skill in the art at the time of the invention to apply the teaching of RFC 3550 above to the system

disclosed by Elliott in view of Szabo for the benefit of calculating PLR using formula

shown above for gaining a better understanding of network performance status.

As to **claim 20**, Elliott in view of Szabo discloses claim 19, Elliott further

discloses the traffic processing (including new call setup) depends on QoS parameters,

including packet loss performance ([1081]);

Elliott does not explicitly disclose the new call is accepted if PLR is below a given

threshold;

however, PLR is just one commonly used QoS parameter;

therefore, it would have been obvious to a person of ordinary skill in the art at the

time of the invention to a new call is accepted if PLR that is (calculated by the third

circuit, CPU) is below a given threshold for the benefit of providing reliable network

service for users.

As to **claim 21**, Elliott in view of Szabo and RFC 3550 discloses of claim 19,

Elliott further discloses the third circuit compares the packet loss ratio;

Elliott does not explicitly disclose the new call is accepted using a reduced

bandwidth if PRL is between given low threshold and the upper threshold;

however, PRL is commonly used QoS parameter ([1081]); and Elliott also

teaches providing different network services depend on QoS parameters, such as delay

and packet loss information ([1088]);

therefore, it would have been obvious to a person of ordinary skill in the art at the

time of the invention to a new call is accepted using a reduced bandwidth if PRL that is

(calculated by the third circuit, CPU) is between given low threshold and the upper

threshold for the benefit of providing reliable network service for users.

As to **claim 22**, Elliott in view of Szabo and RFC 3550 discloses claim 19.

Elliott does not explicitly disclose the new voice call is accepted if PLR is below a

given threshold;

however, PLR is commonly used as a QoS parameter ([1081]);

therefore, it would have been obvious to a person of ordinary skill in the art at the

time of the invention to a new call is blocked if PLR that is (calculated by the third circuit,

CPU) is above the upper threshold for the benefit of protecting normal network

operation.

As to **claim 24**, Elliott in view of Szabo discloses claim 1, but is silent on the

information indicative of the quality of service of voice calls being transmitted from the

first gateway to the second gateway comprises a plurality of performance reports

associated with the voice calls, wherein determining the congestion status parameters

of the network paths comprises: determining, for each of the performance reports, one

of the network paths with which the performance report is associated; and determining,

for each of the network paths, the congestion status parameter of the network path

using at least a portion of the performance reports determined to be associated with the

network path.

RFC 3550 discloses the information indicative of the quality of service of voice

calls being transmitted from the first gateway to the second gateway comprises a

plurality of performance reports associated with the voice calls ("sender report (SR) and

receiver report (RR)", 1st paragraph of Section 6.4, page 35), wherein determining the

congestion status parameters of the network paths comprises: determining, for each of

the performance reports, one of the network paths with which the performance report is

associated (SSRC_1 ... SSRC_n in SR shown in SR packet in Section 6.4.1 of page 36

shows the network path associated with the report); and determining, for each of the

network paths, the congestion status parameter of the network path using at least a

portion of the performance reports determined to be associated with the network path

(one of congestion status parameters "fraction lost", "cumulative number of packets

lost", and "delay since last SR (DLSR)", 1st paragraph of Section 6.4, page 35).

     Therefore, it would have been obvious to a person of ordinary skill in the art at

the time of the invention to apply the teaching of RFC 3550 above to the system

disclosed by Elliott in view of Szabo in order to reduce the network congestion.

     As to **claim 25**, Elliott in view of Szabo discloses claim 1, but does not

specifically discloses the information indicative of the quality of service of voice calls

being transmitted from the first gateway to the second gateway comprises a plurality of

performance reports associated with the voice calls, wherein determining the

congestion status parameters of the network paths comprises: for each of at least one

of the network paths: selecting only a subset of the performance reports associated with

the network path; and determining the congestion status parameter of the network path

using the selected subset of the performance reports associated with the network path.

     RFC 3550 discloses the information indicative of the quality of service of voice

calls being transmitted from the first gateway to the second gateway comprises a

plurality of performance reports associated with the voice calls ("sender report (SR) and
receiver report (RR)", 1st paragraph of Section 6.4, page 35), wherein determining the
congestion status parameters of the network paths comprises: for each of at least one
of the network paths (each report is for a path): selecting only a subset of the
performance reports associated with the network path (congestion status parameters
"fraction lost", "cumulative number of packets lost", and "delay since last SR (DLSR)",
1st paragraph of Section 6.4, page 35, are a subset of the report); and determining the
congestion status parameter of the network path using the selected subset of the
performance reports associated with the network path (congestion status parameters
"fraction lost", "cumulative number of packets lost", and "delay since last SR (DLSR)",
1st paragraph of Section 6.4.1, page 36, are a subset of the report).

    Therefore, it would have been obvious to a person of ordinary skill in the art at
the time of the invention to apply the teaching of RFC 3550 above to the system
disclosed by Elliott in view of Szabo in order to reduce the network congestion.

    As to **claim 27**, Elliott in view of Szabo discloses claim 26, but is silent on for
each of at least one of the network paths, the call control logic is updated using the
congestion status parameter for the network path periodically.

    RFC 3350 discloses for each of at least one of the network paths, the call control
logic is updated using the congestion status parameter (one of congestion status
parameters "fraction lost", "cumulative number of packets lost", and "delay since last SR
(DLSR)", 1st paragraph of Section 6.4, page 35,) for the network path periodically ("each

periodically transmitted compound RTCP packet MUST include a report packet", 1st

paragraph of page 22, Section 6.1).

Therefore, it would have been obvious to a person of ordinary skill in the art at

the time of the invention to apply the teaching of RFC 3550 above to the system

disclosed by Elliott in view of Szabo in order to reduce the network congestion.

As to **claim 28**, Elliott in view of Szabo discloses claim 26, wherein, for each of at

least one of the network paths, the call control logic is updated using the congestion

status parameter for the network path on an exception reporting basis ("Sender and

Receiver Reports", Section 6.4 page 35-45; note that Sender and Receiver Reports are

exception reporting because they provide exception information like "fraction lost",

"cumulative number of packets lost").

Therefore, it would have been obvious to a person of ordinary skill in the art at

the time of the invention to apply the teaching of RFC 3550 above to the system

disclosed by Elliott in view of Szabo in order to obtaining useful network information.

As to **claim 29**, Elliott in view of Szabo discloses claim 1, Elliott further discloses

the first gateway comprise a plurality of ports (FIG. 1 shows each gateway 108 and 110

has a plurality of ports) associated with the respective plurality of network paths (FIG. 1

shows plurality of network paths via Data network 112), for each network path: for each

of at least one voice call being transmitted from the first gateway to the second gateway

via the network path (as shown in FIG. 1).

Elliott is silent on, computing a congestion status value associated with the voice

call using the obtained information associated with the voice call; and determining the

congestion status parameter of the network path using the at least one congestion

status value computed for the network path.

RFC 3550 discloses computing a congestion status value associated with the

voice call using the obtained information associated with the voice call; and determining

the congestion status parameter of the network path using the at least one congestion

status value computed for the network path (each voice call has a path and is

associated with the specific report that includes congestion status parameters "fraction

lost", "cumulative number of packets lost", and "delay since last SR (DLSR)", 1st

paragraph of Section 6.4.1, page 36).

Therefore, it would have been obvious to a person of ordinary skill in the art at

the time of the invention to apply the teaching of RFC 3550 above to the system

disclosed by Elliott in view of Szabo in order to obtaining useful network information.

As to **claim 30**, Elliott in view of Szabo claim 29, but is silent on the congestion

status parameter for the network path is determined by selecting the congestion status

value computed for the network path that is indicative of the greatest amount of

congestion on the network path

RFC 3550 discloses the congestion status parameter for the network path is

determined by selecting the congestion status value computed for the network path that

is indicative of the greatest amount of congestion on the network path (each voice call

has a path and is associated with the specific report that includes congestion status

parameters "fraction lost", "cumulative number of packets lost", and "delay since last SR

(DLSR)", 1st paragraph of Section 6.4.1, page 36; e.g., select the greatest amount of delay because all other delay value is irrelevant anymore).

Therefore, it would have been obvious to a person of ordinary skill in the art at the time of the invention to apply the teaching of RFC 3550 above to the system disclosed by Elliott in view of Szabo in order to obtaining useful network information.

As to **claim 31**, Elliott in view of Szabo claim 1, but is silent on the determination as to whether to accept the new call into the network at the first gateway is performed using all of the network congestion parameters.

RFC 3550 discloses the determination as to whether to accept the new call into the network at the first gateway is performed using all of the network congestion parameters (each voice call has a path and is associated with the specific report that includes congestion status parameters "fraction lost", "cumulative number of packets lost", and "delay since last SR (DLSR)", 1st paragraph of Section 6.4.1, page 36; all these parameters can be used for determination as to whether to accept the new call into the network).

Therefore, it would have been obvious to a person of ordinary skill in the art at the time of the invention to apply the teaching of RFC 3550 above to the system disclosed by Elliott in view of Szabo in order to obtaining useful network information.

As to **claim 32**, Elliott in view of Szabo and RFC 3550 discloses claim 31, Elliott further discloses on accepting the new call into the IP network at the first gateway for transmission toward the second gateway via one of the network paths (as disclosed by the parent claims 1; note that the new call is always accepted via one the network

paths), wherein the one of the network paths for the new call is the one of the network

paths having the associated congestion status parameter indicative of the least amount

of congestion (as disclosed by the parent claims 1: claim 1 discloses congestion

information for different paths among which there is one path that has the least amount

of congestion).

9.   **Claims 2** and **13** are rejected under 35 U.S.C. 103(a) as being unpatentable over

Elliott in view of Szabo, further in view of Watt (US Patent number 5781532, hereinafter

Watt).

As to **claim 2**, Elliott in view of Szabo discloses the method of claim 23, but are

silent on wherein new call may be accepted at a reduced bandwidth in the case of said

parameter exceeding a lower threshold.

In the same field of endeavor, Watt discloses adjusting transmission rate in

response to a "mild" congestion state, as indicated by the value packet loss ratio

exceeding a lower threshold but below an upper threshold, "*adjusting* the transmission

rate in response to the detection of said congestion … when the detected congestion

exceeds a predetermined *mild congestion threshold*", from line 21 of col. 7 to line 3 of

col. 8).

Therefore, it would have been obvious to a person of ordinary skill in the art at

the time of the invention to accept a new call at a reduced bandwidth in order to fully

take advantage of network resource.

As to **claim 13**, Elliott in view of Szabo and Watt discloses the method of claim 2,

Elliott disclose further discloses wherein the bandwidth of a newly accepted call is

reduced by activating the characteristic of silence suppression for said newly accepted

voice call (silence suppression activation timer, table 147 in Page 85).

10.   **Claims 11-12** are rejected under 35 U.S.C. 103(a) as being unpatentable over

Elliott in view of Szabo and Watt, further in view of RFC 3550.

As to **claim 11**, Elliott in view of Szabo and Watt discloses claim 2, Elliott further

discloses using different encoders (CODECs, such as ones supporting G.711, G. 726,

and G.728 in [1004]) to handle different connections with different bandwidth ([1004]);

which include the case of using 2 different encoders to handle 2 different kinds of calls

that have different bandwidth.

As to **claim 12**, Elliott in view of RFC 3550 and Szabo discloses the method of

claim 2, but are **silent on** wherein the bandwidth of a newly accepted call is reduced by

increasing the packet size for said newly accepted voice call.

However, for a given amount of data, increasing the packet size will decrease the

overhead caused by packet header therefore reduce the required bandwidth for the call,

(this is demonstrated by efficiency factor e=(packet_size)/(packet_size+header_size);

for a fixed header size, the larger is the packet_size, the larger is e);

Therefore, it would have been obvious to a person of ordinary skill in the art at

the time of the invention to increase the packet size so as to decrease the required

bandwidth for the call for the benefit of saving bandwidth resource.

11.   **Claims 16-17** are rejected under 35 U.S.C. 103(a) as being unpatentable over

Elliott in view of RFC 3550 and Szabo, further in view of Hooper et al (US 20040252686

A1) (hereinafter **Hooper**).

As to **claim 16**, Elliott in view of Szabo discloses the apparatus of claim 14, but are silent on said second circuit is at least one strongarm card.

However, the second circuit is simply a CPU card (such as CPU card of Soft Switch 204, FIG. 2B) that implement the logic of receiving QoS information associated with voice calls, and strongarm is a popular CPU that is commonly used in the CPU cards as disclosed by Hooper ("the core processor implemented as a Strong Arm architecture", [0010]).

Therefore, it would have been obvious to a person of ordinary skill in the art at the time of the invention to use strongarm CPU card as the circuit disclosed by Elliott for the benefit of cost saving.

As to **claim 17**, Elliott in view of Szabo and Hooper discloses the apparatus of claim 16, Elliott further discloses the CPU card (with strongarm CPU) is connected to the Ethernet card via a host CPU circuit (CPU card of Soft Switch 204 is connected to Ethernet switches 332, [0568], line 1-5 and FIG. 2B).

Response to Amendments/Remarks

12.   Applicant's arguments and all other documents filed on 11/22/10 have been fully considered but are not moot because all claims have been amended to which new ground rejections are made. For details See the rejection addressing new amendment above.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Jianye Wu whose telephone number is (571)270-1665.

The examiner can normally be reached on Monday to Thursday, 8am to 7pm. If

attempts to reach the examiner by telephone are unsuccessful, the examiner's

supervisor, Seema Rao can be reached on (571)272-3174. The fax phone number for

the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent

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have questions on access to the Private PAIR system, contact the Electronic Business

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800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/Jianye Wu/
Examiner, Art Unit 2462